L Number	Hits	Search Text	DB	Time stamp
1	2081	(time adj domain) and (hrtf or (head adr related adj transfer adj function))	USPAT	2002/09/16 11:03
2	0	(time adj domain) and (covariance adj data adj matrix)	USPAT	2002/09/16 11:04
3	12	((time adj domain) and (hrtf or (head adr related adj transfer adj function))) and (covariance with matrix)	USPAT	2002/09/16 11:04
5	1	((time adj domain) and (hrtf or (head adj related adj transfer adj function))) and (covariance with matrix)	USPAT	2002/09/16 11:04
4	42	(time adj domain) and (hrtf or (head adj related adj transfer adj function))	USPAT	2002/09/16 11:31
6	1	((time adj domain) and (hrtf or (head adj related adj transfer adj function))) and (eigen or eigenfilter$3)	USPAT	2002/09/16 11:19
7	7	(time adj domain) and (hrtf or (head adj related adj transfer adj function))	EPO; JPO; DERWENT; IBM_TDB	2002/09/16 11:31
8	0	((time adj domain) and (hrtf or (head adj related adj transfer adj function))) and (eigen or eigenfilter$3)	EPO; JPO; DERWENT; IBM_TDB	2002/09/16 11:31
9	1	(hrtf or (head adj related adj transfer adj function)) and (eigen or eigenfilter$3)	USPAT	2002/09/16 11:32

L Number	Hits	Search Text	DB	Time stamp
1	78	(impulse adj response) and (hrtf or (head adj related adj transfer adj function))	USPAT	2002/09/16 12:23
2	1	((impulse adj response) and (hrtf or (head adj related adj transfer adj function))) and (eigen or eigenfilter$3)	USPAT	2002/09/16 12:26
5	88	(head adj related) and ((impulse adj response) or (time adj domain))	USPAT	2002/09/16 12:26
6	1	((head adj related) and ((impulse adj response) or (time adj domain))) and (eigen or eigenfilter$3)	USPAT	2002/09/16 12:30
7	1	6144747.pn.	USPAT	2002/09/16 12:31

US-PAT-NO: 6144747

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TITLE: Head mounted surround sound system

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Some present research that is going on in the multi-dimensional sound system field is that for developing a multisensory "virtual environment" work station (VIEW) for use in space station teleoperation, tele-presence and automation activities. The auditory requirements for this project led to the prototyping of a binaural signal processor for converting generated or recorded sounds into binaural signals. Researchers measured a subject's pinna responses as a function of azimuth and elevation and arrived at pure head related transfer functions (HRTFs) using Fast Fourier Transform techniques. These HRTFs were implemented in a Digital Signal Processing (DSP) device which allowed the user to apply direction dependent equalization to an incoming signal. By establishing the proper relationship between the I'D, the Interaural Level Difference (ILD), and the HRTF, experimenters were able to synthesize free field stimuli and present this over headphones. Motion trajectories and static locations that represented greater resolution of HRTFs than measured were arrived at through interpolation. However, this system had some problems with front-back reversals.

Referring now to FIG. 16, there is illustrated a block diagram of a system for providing real-time convolution in order to convolve the impulse response of a given environment, such as a theater. In addition to providing the surround sound system, it is also desirable to provide the surround sound system in conjunction with the acoustics of a given theater. Some theaters are specifically designed to facilitate the use of surround sound and they actually enhance the original surround sound of the audio track. This convolution may be performed directly in the computer in the time domain which, however, is a slow process unless some type of special computer architecture is utilized. Normally, convolution is usually in the form of its frequency domain equivalence since the Fourier transformation of the audio signal and impulse response, followed by the multiplication and inverse fast Fourier transformation of the result are faster than direct convolution. This method can be implemented with software or hardware. This type of convolution is

often performed using a computer coupled to an array processor, the advantage being that input signals and room impulse responses may be arbitrarily long, limited only by the computer hard disk space. However, the disadvantage of the system is that the processing time of the impulse response is comparatively long. The present invention utilizes a digital signal processor (DSP) as a signal processor to provide a digital filter that can convolve a multiple channel impulse response and a predetermined sampling frequency in real time with only a few seconds of delay. One type of real-time convolver is that manufactured by Signal Logic Inc., which allows the user to perform either mono or binaural audible simulations ("auralizations") in real-time using off-the-shelf DSP/analog boards and multi-media boards. The filter inputs are typically any impulse response.